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The Lincoln Low-Rate Vocoder: A 1200/2400 bps LPC-10 Voice Terminal

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LEXINGTON, MASSACHUSETTS

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FOR THE COMMANDER

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THE LINCOLN LOW-RATE VOCODER: A 1200/2400 bps LPC-10 VOICE TERMINAL

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Group 24

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ABSTRACT

The following is a description of a low data rate voice terminal. The unit uses an LPC-10 algorithm which provides a 2400 bps NATO standard compatible mode and a 1200 bps frame-fill mode. The unit has an RS-449 digital interface with MIL-STD-188-114 levels. The user operates the unit from a telephone instrument with a PTT handset. The user interface models an augmented combination of an FM transceiver and a telephone to inform the user of the state of the data communications equipment (DCE) in a natural manner. These protocols cover both full and half duplex channels. The unit exhibits a DRT (intelligibility) score of 91.1% at 2400 bps and 88.7% at 1200 bps in benign acoustic environments.

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I. INTRODUCTION

The government standard LPC-10 vocoder algorithm [1] and its associated NATO standard bitstream [2] have facilitated 2400 bps speech communication. This report describes an extension of a compatible 2400 bps vocoder algorithm which halves the data rate with only a slight loss of intelligibility. The 1200 bps algorithm builds on the underlying 2400 bps vocoder by using the original analyzer, synthesizer, and modem routines. The basic algorithm used is frame fill, a controlled alternate frame interpolation scheme. The unit includes a standard digital interface with a serial bitstream and control signals to allow control of the DCE from the vocoder unit. The unit also incorporates a human interface to inform the user of the state of the DCE and vocoder unit.

This report is both a description of the 1200 bps algorithm and the Lincoln Low-Rate Vocoder unit itself. The first portion describes the algorithm, coding, and bitstream in sufficient detail that the reader can design a compatible 1200 bps vocoder. The second portion is a user guide to the unit. It includes descriptions of the RS-449 interface, the human interface and all controls and indicators.

II. GENERAL SPECIFICATIONS

<u>Algorithm</u>:

LPC-10:

2400 bps: NATO standard-compatible.
1200 bps: frame-fill.

Intelligibility:

Quiet background, dynamic microphone, 3 male talker DRT:

2400 bps: 0 ber	91.1%	(std err: .66%)
1200 bps: 0 ber	88.7%	(.80%)
.1% ber:	89.8%	(.57%)
1.0% ber:	84.3%	(1.07%)

Digital Interface:

RS-449 with MIL-STD-188-114 levels.
Serial data stream.
DB-37 connector.

Audio Interface:

4 kHz audio bandwidth.
Telephone instrument with noise cancelling microphone.
Auxiliary line level I/O.
BNC connectors.

Human Interface:

User Model:
Combination of telephone and FM transceiver.
Telephone instrument:
PTT and off-hook controls.
Transmit and receive lights, buzzer.
"State indication sounds".
Sidetone with distortion to indicate overload.

Delay:

2400 bps: 110 ms.
1200 bps: 180 ms.

Implementation:

TTL bit-slice programmable processor.

Size:

3.5" h. x 17" w. x 18" d. (19" rack mountable).

Weight:

22 lbs.

Power:

120 VAC., 50-60 Hz., 70 Watts.

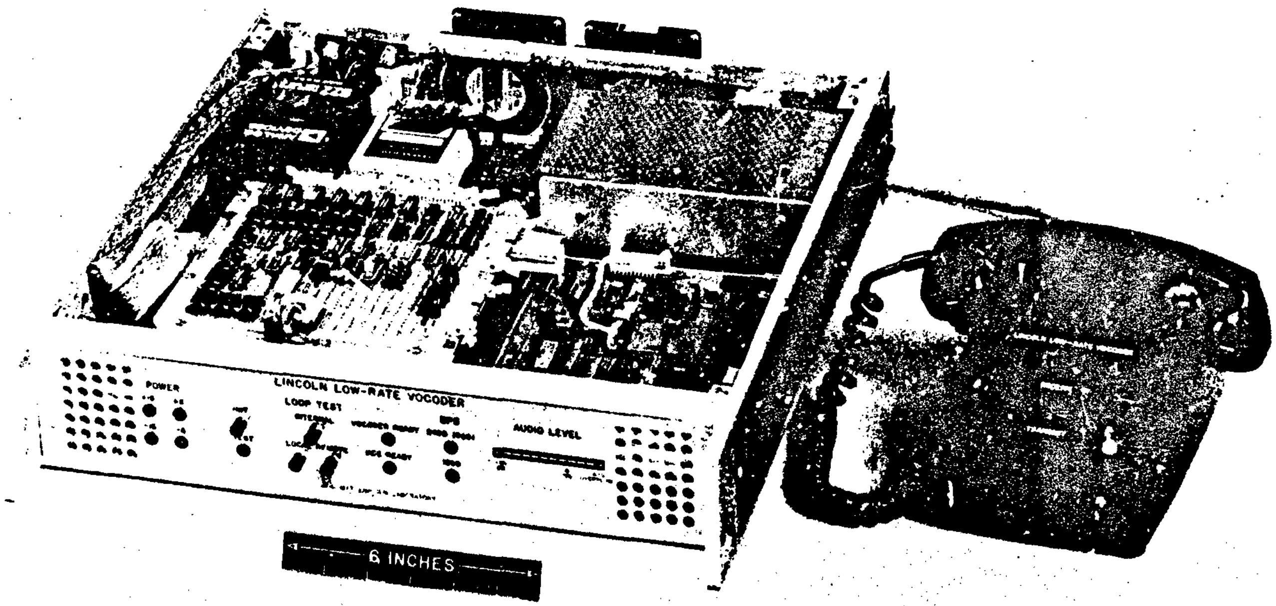

Fig. 1. The Lincoln low-rate vocoder.

III. THE 1200 BPS ALGORITHM

The 1200 bps mode is achieved by alternate frame fill [3,4,5]. In this scheme, the transmitter replaces alternate data transmission frames by a few control bits and the receiver reconstructs the missing frames from the received frames according to a strategy specified by the control bits. An equal number of bits are removed from the coded parameters in the transmitted frame to provide an exact halving of the data rate. The implementation used here assumes the existence of a government standard bitstream system and embeds the frame-fill routines. The vocoder algorithm and modem routines are unchanged. The vocoder frame rate is unchanged, but the modem frame rate is halved resulting in two vocoder frames per transmission frame. The frame-fill algorithm uses parameters which look like the 2400 bps parameters to the modem routines but are coded differently for the 1200 bps mode.

The parameter notation used in this section is the following: a numerical suffix is a reverse time order frame index. Thus the parameter x for the frames of interest is x4, x3, x2, x1, x0 in temporal order. The even frames are transmitted and frame 1 is the frame (currently) to be filled.

Frame fill of the pitch and voicing parameters requires transmission of only a voicing bit for the filled frame. (The 2400 bps system used here, unlike the government algorithm [1,2], has no voicing transition states. All received transition states are mapped to unvoiced.) The pitch and voicing are inferred at the receiver from the following table from [4]:

Voicing Sequence qp2,vb1,qp0	Inferred Pitch/voicing
u u u	qp1 = u
u u v	qp1 = u
u v u	qp1 = u
u v v	qp1 = qp0
v u u	qp1 = u
v u v	qp1 = (qp2+qp0)/2
v v u	qp1 = qp2
v v v	qp1 = (qp2+qp0)/2

where: pn = pitch for frame n
vb1 = voicing control bit for frame 1
v = voiced
u = unvoiced
qx = quantized x (coding and decoding quantization) i.e., the value of x found at the receiver.

The inferred pitch value for the vuv sequence is a judgment based upon the assumption that this sequence represents a voicing detection or data transmission error [6]. (An equally valid case could also be made for an unvoiced inference for this sequence.) Our experience shows the value used here to work well. Since this decision does not affect the transmitted bit stream, the individual units can use either choice. (This sequence occurs rather infrequently which suggests that the choice is not very important.)

The filled-frame amplitude (rms of the frame energy) is the closest of four values to the actual value according to an L_1 norm of the log amplitudes (i.e., the closest value on a db scale). Since the receiver can only have choices generated from quantized values it has received during the even frames, these quantized values are generated in the transmitter for comparison to the unquantized value from the analyzer. A computationally efficient way of approximating this "L_1 norm of the log amplitudes" is to first encode the amplitudes using the 6 bit rms coding

table (Appendix A), perform the 5 bit quantizations (Appendix A) on the candidate values, and perform the search for the minimum absolute distance. The amplitude values for the transmitted (even) frames will be coded to 5 bits. These are the four amplitude choices:

Candidate for ca1	Code (control) value	
c5a0	0	(e1=0, e0=0)
(c5a0+c5a2)/2	1	(e1=0, e0=1)
c5a2	2	(e1=1, e0=0)
min(max(c5a0,c5a2)+6,63)	3	(e1=1, e0=1)

desired value = c6a1

where cx = coded x
c5x = x coded to 5 bits (in 6 bits)
c6x = x coded to 6 bits
an = amplitude for frame n
e1,e0 = amplitude control bits

The last candidate is intended to aid representation of short amplitude bursts which are surrounded by lower amplitude regions. It amounts to the greater of the adjacent frame amplitudes plus 4.6 dB subject to the limitation that it not exceed the range of the coding table. The metric is chosen to mimic human amplitude perception and the second choice is chosen to approximate a perceptual midpoint between the first and third choices.

The filled spectrum (K vector) is one of four choices as determined by a weighted L_2 norm of the LAR's (log-area ratios):

$$d(J,K) = \sum_{i=1}^{10} \{w[i] * (LAR(J[i])-LAR(K[i]))^2\}$$

where x[i] = ith component of x
w[1] = 1
w[i] = 1-(i-2)/16 $\quad 2 \leq i \leq 10$
J,K = K vectors
LAR(x) = log((1+x)/(1-x))

The LAR is approximated by a 256 entry table spanning k= -.98 to k=.98 in uniform increments in LAR (Appendix B). This distance measure was developed for use in a vector quantizer and has been shown empirically to perform well in that environment [7]. It represents an attempt to match human spectral (envelope) perception in a K-based domain. The four choices are as follows:

Candidate for qK1	Code (control) value	
qK0	0	(s1=0, s0=0)
LARavg(qK0,qK2)	1	(s1=0, s0=1)
qK2	2	(s1=1, s0=0)
qK4	3	(s1=1, s0=1)

desired value = K1

where	qx	= quantized x
	Kn	= K vector for frame n
	s1,s0	= spectrum control bits
	LARavg(X,Y)	= invLAR((LAR(X)+LAR(Y))/2) (componentwise for the vectors X and Y)
	invLAR(x)	= inverse LAR(x)

The preceding defines an algorithm and a set of five control bits to control the interpolation. The bitstream must now be modified to include these control bits and halve the total data rate. The convention chosen is based upon the government standard bitstream and coding tables. First, the modem clock is halved from 2400 to 1200 bps. Secondly, five of the parameters are coded in one fewer bit than the government standard:

Field	Gov't Std	1200 bps	Transmitted "Parameter"
Sync	1	=	=
Fill ctrl			
v/u		1	
amplitude		2	
spectrum		2	
Pitch	7	6	vbl*64 + cp0
Amplitude	5	=	= (ca0)
K0[1]	5	=	=
K0[2]	5	=	=
K0[3]	5	=	=
K0[4]	5	4	e1*16 + cK0[4]
K0[5]	4	=	=
K0[6]	4	=	=
K0[7]	4	3	e0*8 + cK0[7]
K0[8]	4	3	s1*8 + cK0[8]
K0[9]	3	2	s0*4 + cK0[9]
K0[10]	2	=	=
	54 bits	54 bits	

where x[i] = ith component of the vector x
"=" = same as gov't std

The parameters which are coded in the same number of bits in both modes use the same coding tables (or algorithm) as the government standard [1,2]. The K parameters which are coded in one fewer bit are coded using the tables in Appendix C. The transmitted "parameters" for the 1200 mode are a combination of the (1200 bps) coded value and one of the fill control bits in the (2400 bps) msb position. Thus these new "parameters" can be given to the 2400 bps dispersion and modem routines in the transmitter and received from the 2400 bps dispersion and modem routines in the receiver. Several assumptions are made to allow this overlay:

(1) All 10 K's are always transmitted.

(2) The dispersion routines assume the frame to be voiced.

(3) No forward error correction (FEC) is used.

The pitch is coded to 6 bits by the following equation:

$$cp0 = \begin{cases} 0 & \text{if } p0 \text{ unvoiced} \\ p0-19 & 20 <= p0 < 40 \\ p0/2+1 & 40 <= p0 < 80 \\ p0/4+21 & 80 <= p0 <= 156 \end{cases}$$

using truncation arithmetic. Decoding is accomplished by inverting the above equation. cp0 can thus range from 0 to 60 inclusive. If any of the remaining values (61, 62, or 63) is received, it is assumed to be the result of an error and is decoded as unvoiced. (This turns out to be a rather frequent error--reception of all ones is the quiescent state of the RS-449 interface.) These pitch quantization values are identical to the government standard quantization values--only the error correction and voicing transition state have been removed.

IV. THE DIGITAL INTERFACE

The digital interface is a synchronous RS-449 interface [8] with MIL-STD-188-114 levels [9]. All lines which may be balanced as provided in [8] are implemented as such. The following table describes the interface implemented in the vocoder unit.

Pin	Circuit	RS-449 Description	Function	Vocoder Circuit
1	Shield	-		
2	SI #	Signaling rate ind.	1200/2400 bps **	r1
3	-			
4,22	SD +	Send data	same	t2
5,23	ST	Send timing	Transmit clock	r2
6,24	RD	Receive data	same	r2
7,25	RS + *	Request to send	PTT **	t2
8,26	RT	Receive timing	Receive clock	r2
9,27	CS *	Clear to send	Clear to talk	r2
10	LL +	Local loopback	Request local loop test	t1
11,29	DM *	Data mode	Interface unit ready	r2
12,30	TR + *	Terminal Ready	Vocoder ready	t2
13,31	RR *	Receiver ready	Data being received	r2
14	RL +	Remote loopback	Request remote loop test	t1
15	IC	Incoming call	unused	
16	SF/SR+#	Sel freq/Sig rate sel	unused	on d1
17,35	TT +	Terminal timing	unused	
18	TM *	Test mode	Loop test granted	r1
19	SG	Signal ground	same	
20	RC	Rece've common	Reference for unbal lines	
21	-			
28	IS + *	Terminal in service	power on	on f1
32	SS +	Select standby	unused	off d1
33	SQ #	Signal quality	Received data errors	r1
34	NS +	New signal	unused	off d1
36	SB	Standby indicator	unused	
37	SC +	Send common	Reference for unbal lines	

+ = to DCE
# = fail safe on rec
* = fail safe off rec
DCE = data communications equipment

** = see "back panel controls"

t=transmitter
d=dummy trans
f=fixed trans
r=receiver
1=unbalanced
2=balanced

Notes:

1. These notes do ot constitute a full summary of RS-449 and MIL-STD-188-114. The eader is referred to [8] and [9] for details.

2. The balanced line names are abbreviated: XY is actually XYA and XYB.

3. DB-37 connector (male contacts, female shell).

4. MIL-STD-188-114 levels [9]:
   -4V = data 1 = off = mark
   +4V = data 0 = on = space.

5. DM, TR, and IS must be on for normal operation.

6. SI (from DCE) controls the data rate: on=2400 bps, off=1200 bps. (ST and RT, if supplied by the DCE, must agree.) (See "back-panel option switch".)

7. TT can supply either or both of ST and RT (modem clocks).

8. Transmit protocols:
   PTT (from handset) sets RS. (See "back-panel controls".)
   When ready, the DCE sets CS which turns the vocoder transmitter on.

9. Receive protocols:
   RR (from DCE) causes the vocoder receiver to operate.
   SQ (from DCE) on causes normal operation.
   Off reduces the output level by 6 dB to indicate unreliable data.

10. Test modes:
    Internal loop: cancels IS and sets internal loop and "test" light.
    Local loop: sets LL.
    Remote loop: sets RL.
    TM (from DCE) indicates acceptance of LL or RL and lights "test" light.
    Any test mode will cancel the sidetone. (See Human interface and back panel controls.)

11. Grounds:
    SG: connection through 100 Ω (1/2 W) to vocoder chassis.
    This resistor may burn if the chassis potentials differ. See [8].
    SC: reference for unbalanced line receivers in DCE.
    RC: reference for unbalanced line receivers in vocoder.
    Shield: no connection in vocoder. See [8].

12. Signal terminations:
    Vocoder transmitters:
      Series 50 Ω resistors on all balanced lines.
      Slew rate limiting on unbalanced lines.
      Dummy transmitters are 100 Ω series resistors to 5 volts.
    Vocoder receivers:
      300 Ω parallel on RD, RT, and ST.
      Fail-safe receivers: 5 kΩ to 5 volts.
        Defaults in table.
        (The fail-safe receivers will detect only open circuits.)

13. The audio level meter indicates missing RT or ST by special patterns. See "The Front Panel."

V. THE USER INTERFACE

A user interface has been designed to present the state of the vocoder and DCE to the user in a natural manner. The system looks to the user like an augmented combination of a telephone and an FM radio unit. Since the vocoder is a full-duplex unit, the protocols of this user model are intended to cover both full and half-duplex operation. The following list of protocols for the human interface assumes the back panel option switch is set as recommended for a typical setup (see back panel option switch):

(1) If either the vocoder or the DCE are not operational (RS-449 TR and DM), the telephone instrument will appear to be dead.

(2) To transmit:

a. Lift handset from the cradle. If no signal is being received, a soft white noise should be heard.

b. Press the PTT (push-to-talk) button on the handset. A tone should be heard. The tone indicates that a request to transmit (RS-449 RS) has been issued to the DCE.

c. When the tone ceases and the transmit light on the telephone instrument lights, the DCE has replied with a "clear to transmit" (RS-449 CS) and the user may speak.

d. The user may observe his speaking level on the front panel audio level indicator (see "Front Panel"). A noticeable distortion in the sidetone indicates overload and may be used to adjust the speaking level of a user who cannot see the front panel. The best level is just below overload. Occasional overload on speech peaks should cause very little degradation.

e. The soft white noise will reappear (unless the receiver is operating) and the "transmit" light will extinguish when the PTT button is released.

f. If the "clear to transmit" (RS-449 CS) signal is removed for any reason (such as transmitter failure) during the transmission, the single tone will reappear and the transmit light will extinguish.

g. If the transmit clock (RS-449 ST) is not present during an attempt to transmit, a raucous two tone signal will be heard and an error display will appear on the audio level display (see "Front Panel").

h. If the back panel option switches "fd24" (2400 bps) or "fd12" (1200 bps) are set, steps a and b will merge and the user need not depress the PTT button since the PTT function will be activated by the telephone hook switch. This is used only for a full-duplex channel.

i. If the "sidetone" switch (Back Panel) is on, sidetone will be present during transmission unless the "test" light is lit (RS-449 TM or "Internal Loop").

(3) To receive:

a. An incoming signal (indicated by RS-449 RR) will cause the (on hook) telephone instrument to buzz and the "receive" light will flash.

b. When the handset is lifted from cradle, the buzzer will cease and the "receive" light will become steady.

c. When the incoming signal terminates, the "receive" light will extinguish and (unless transmitting) soft white noise will appear in the handset.

d. If there is a high probability of data errors (RS-449 SQ) or the vocoder receiver is out of (input data stream) synchronization, the receive volume will be reduced by 6 db.

e. If the receive clock (RS-449 RT) is not present during an attempt to receive and the unit is not transmitting, a raucous two tone signal will be heard and an error display will appear on the front panel audio level display.

In summary, the protocols of the human interface are more complicated to analyze than to use. If the RS-449 interface is properly utilized, the user will just answer the phone and listen when it rings (buzzes), and talk after pressing the PTT button. The unit will allow simultaneous talking and listening over a full-duplex channel, but only one at a time over a half-duplex channel.

VI. THE FRONT PANEL

The front panel contains a number of indicators and four controls. The indicators are:

4 power lights (red and green):
+15, +5, -15, and -5.
All must be lit.

"test" (red):
Indicates the unit is in a loop test mode.
Pressing the "internal" test button should always set "test".
Pressing "local" or "remote" will set "test" only if the DCE (or DCE, channel, and remote modem) grant the requested test condition.

If "test" is lit and no button is pressed, the internal loop switch or RS-449 TM is set. See "Internal Controls" and "The Digital Interface".

"vocoder ready" (green):
Indicates the vocoder unit is operational and RS-449 TR is on. These should always be on when the unit is powered. (If not on, the "init" button should set the light.) They must be on for the vocoder to transmit or receive speech.

"DCE ready" (green):
Indicates the DCE (indicated by RS-449 DM) is ready for operation. Must be on for the vocoder to transmit or receive speech.

"BPS" "2400 (DoD)" (red) and "1200" (green):
Indicates data rate as set by RS-449 SI or the back panel option switch.

"Audio Level" (green and red bar graph):

Indicates the input audio level when transmitting. The unit is calibrated in 6 db steps. If the rightmost red light is lit, the A/D converter or the algorithm is overloaded. The best audio level is just below overload, but occasional overload on speech peaks should do no harm.

This indicator also indicates missing modem clocks with special patterns (g=green, r=red, and -=off):

```
ST missing:   g------rrr
RT missing:   -g-----rrr
both missing: gg-----rrr
```

The controls are:

"init":

Initializes the unit. This should not generally be required since the unit auto-initializes on power-up.

"Loop Test":

"Internal":

Sets an internal digital loop and sets the "test" light. The user should be able to speak into the microphone and hear his vocoded (and delayed) speech in the earpiece. This test condition will always be granted.

"Local":

Requests a loop test through the DCE (RS-449 LL). If granted, the test light will light and the user should be able to hear his vocoded speech as above.

"Remote":

Requests a loop test from the remote modem (RS-449 RL). (This is possible only on a full-duplex channel.) If granted, the test light will light and the user should be able to hear his vocoded speech as above.

VII. THE BACK PANEL

The back panel has several connectors, the power switch, a fuse, and an option control switch:

"115 VAC 50/60 Hz":
Power line cord.

"ON/OFF":
Power switch.

"Fuse":
2A slow-blow, 3AG fuse.

Option switch:
This DIP switch allows the user to control several aspects of the digital and human interfaces.

label	function	left	right
"bps"	use internal data rate selection	internal	external (RS-449 SI)
"int bps"	set internal data rate ("bps"=int only)	2400 bps	1200 bps
"fd24"	full-dux at 2400 bps (take PTT from hook)	on	off
"fd12"	full-dux at 1200 bps (take PTT from hook)	on	off
(unused)			
"buzz"	use buzzer when telephone rings	on	off
"sidetone"	control audio sidetone (automatically off for all test modes)	on	off
(unused)			

A typical setup will be:

label	setting
"bps"	ext
"int bps"	(either)
"fd24"	off
"fd12"	off
(unused)	-
"sw buzz"	on
"sidetone"	on
(unused)	-

The above setup would be used in a half-duplex environment. In a full-duplex environment the user may elect to turn "fd24" and "fd12" on. The act of lifting the handset from the cradle will now key the transmitter and the PTT button need not be used.

"Line In":
Line level audio input. (Isolated ground BNC connector.)

"Line Out":
Line level audio output. (Isolated ground BNC connector.)

"Phone":
Connects to cable to telephone instrument.

"Digital":
DB-37 connector to DCE.

VIII. INTERNAL CONTROLS

The unit has only two classes of internal controls: internal loop and audio gain switches. In general, one should not need to access these controls. Access to these switches is gained by removing the top cover. (WARNING: 120 VAC powerlines exist within the unit.)

"Loop" (toggle switch on small board):
This is an internal switch which locks the "internal loop test" found on the front panel. It should be left off since it will render the unit inoperative for communication over an external data line.

Gains (on large board):
The following controls are DIP switches which control the audio system. The relative gains for each switch are given and all set gains are additive. The switches for which only a setting are given control other portions of the audio system and MUST be set as indicated for proper operation. Switches whose function is not indicated are not used.

"Microphone gain" (location EE1)

switch	function	suggested setting
1	gain of 16	off
2	gain of 8	on
3	gain of 4	off
4	gain of 2	off
5	gain of 1	off

"Earpiece gain" (location FF43)

switch	function	suggested setting
1	gain of 1	off
2	gain of 2	off
3	gain of 4	on
4	gain of 8	off
5	gain of 16	off

"Line in gain" (location EJ1)

switch	function	suggested setting
1	gain of 16	off
2	gain of 8	on
3	gain of 4	off
4	gain of 2	off
5	gain of 1	off
6	MUST BE OFF	OFF

"Line out gain" (location EJ40)

switch	function	suggested setting
1	gain of 1	off
2	gain of 2	off
3	gain of 4	off
4	gain of 8	on
5	gain of 16	on
6	MUST BE OFF	OFF

"Program switches" (DIP switch at location FF1)

These switches are not used in this system.

IX. THE TELEPHONE INSTRUMENT

The telephone instrument used here is specially designed for use with the vocoder unit. It has two controls: a push-to-talk (PTT) button on the handset, and the hook switch which is controlled by hanging up or lifting the handset from the cradle. These controls are the only controls required by the vocoder unit in normal operation. The telephone instrument has

"receive" and "transmit" lights and an internal buzzer. The microphone in the handset is a noise cancelling unit which requires that the user speak very closely for proper operation.

The two control signals enter default states when the telephone instrument is unplugged. The defaults are: off-hook and PTT off. This will allow the vocoder to accept a received signal but not attempt to transmit. (A soft white noise will appear on the "line out" unless a received signal is present. See the "Human Interface".) The unit can be forced to transmit by setting the appropriate back panel switch "fd24" or "fd12". Audio input and output must now occur via the "line in" and "line out" connectors.

X. IMPLEMENTATION

The unit is implemented using a TTL bit-slice programmable processor [10] which has about 100 DIP's on a single universal board. This processor has been repackaged (Fig. 1) and modified in several ways. The audio system has been augmented to provide dual independent inputs and outputs to interface to both the telephone instrument and the line-level I/O. Additional registers were added to both the input and output busses to provide I/O to the various control functions and the RS-449 control lines. The RS-449 interface functions are implemented on a second board containing about 45 DIP's.

XI. ACCESSORIES

Two RS-449 connection boxes are used with these vocoder units for test and demonstration purposes. The first is a cross connection. This unit is used to connect two of the vocoder units together for conversation tests.

The switch is used to set the data rate via the RS-449 SI line. The back panel options should be set as recommended in the "Back Panel" section except that "fd24" and "fd12" may be set for either half or full-duplex. The RS-449 control signals are cross connected such that transmission from either unit will initiate reception in the other unit. The cross connection simulates a full-duplex channel.

The second connection box implements a local loop. This loop connects the transmitter back to the receiver of the same unit. The switch is again used to set the data rate via the RS-449 SI line. The back panel options should again be set as recommended in the "Back Panel" section. This will simulate a half-duplex channel except that both the transmitter and receiver are in the same vocoder unit. The local loop connection box asserts the RS-449 TM line (and lights the test light) to cancel the sidetone to prevent an echo which may be offensive in some tests.

XII. VOCODER EVALUATIONS

The simplest setup for one of these vocoders for speech intelligibility tests such as the DRT is as follows:

(1) Use the local loop connection box. This will simulate an RS-449 interface, setup the loop and data rate and cancel the sidetone.

If the local loop connection box is not used, the data rate may be set and the sidetone must be canceled from the back panel option switch ("sidetone"= off). The user is now responsible for the RS-449 interface.

(2) Set the back panel options as recommended in the "Back Panel" section except set the "fd24" and "fd12" options on.

(3) Disconnect the telephone instrument. This will prevent stray noise pickup in the microphone and set the off-hook condition.

(4) Connect the input speech to the "line in" connector and set the level by the front panel audio level indicator as described in the "Front Panel".

(5) Take the output (vocoded) speech from the "line out" connector.

To perform similar tests using two vocoder units:

(1) Use the cross connection connection box. This will simulate the RS-449 interfaces and set the data rate.

If the cross connection connection box is not used, the data rate may be set from the back panel option switchs. The user is now responsible for the RS-449 interfaces.

(2) Set the back panel options on both units as recommended in the "Back Panel" section except set "fd24" and "fd12" on and "sidetone" off.

(3) Disconnect the telephone instruments from both units. This will prevent stray noise pickup in the microphones and set the off-hook condition.

(4) Connect the input speech to the "line in" connector of the transmitter unit and set the level by the front panel audio level indicator as described in the "Front Panel".

(5) Take the output (vocoded) speech from the "line out" connector of the receive unit.

XIII. DISCUSSION AND CONCLUSIONS

The voice terminal described here shows that high quality digital speech communication is possible at 1200 bps with a simple implementation. (Similar intelligibility systems at lower data rates have been achieved, but the implementations are more complex. See, for instance, [5] or the paper cited in [7].) The human interface has been designed to make the unit

"do the natural thing" with respect to both the channel and the speech and thus make the unit very easy to use. The digital interface provides a sufficient set of control functions to integrate the channel control and status sensing into the human interface.

The 1200 bps frame-fill algorithm has been designed with conversion between 2400 bps (government standard) and 1200 bps as a secondary goal. Where feasible, the government standard compatible codings have been used. Thus, the pitch, amplitude, and six of the K data format conversions are straight forward. (In conversion from 2400 bps to 1200 bps, the uncoded parameter values are not available, so the 2400 bps coded values must be substituted.) The remaining four K conversions will incur an ambiguity and bias on alternate code values during conversion from 2400 bps to 1200 bps (i.e., converting the parameter from an n to an n-1 bit code) due to incompatibilities in the coding tables. These ambiguities occur because half of 2400 bps decoded values are midway between the available 1200 bps decoded values. (The other half of the 2400 bps decoded values are identical to the 1200 bps decoded values and therefore cause no difficulties.) When this ambiguity occurs, use the code (of the two potential 1200 bps codes) corresponding to a decoded value whose magnitude is smallest.

The performance of this algorithm and recently demonstrated efficient implementations of LPC vocoders [11] suggest a very compact secure voice terminal for use over dial-up telephone lines. The 1200 bps data rate is currently in common use for full-duplex digital communication over (2 wire) dial-up telephone lines using inexpensive modems. A 1200 bps

version of the compact vocoder described in [11] is feasible [6] with commercial IC technology. One-chip DES encryption systems and few-chip 1200 bps modems are beginning to appear on the market. Thus, the combination of a compact vocoder, a one-chip encryption system, and a compact telephone modem could implement a full-duplex secure voice terminal which would fit in the telephone body itself.

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Appendix A
Frame Amplitude Coding Table

The amplitude is coded according to the following table:

code-value			
0-0	16-16	32-64	48-270
1-1	17-17	33-70	49-294
2-2	18-18	34-78	50-328
3-3	19-20	35-84	51-352
4-4	20-22	36-92	52-384
5-5	21-24	37-102	53-420
6-6	22-26	38-110	54-460
7-7	23-30	39-120	55-502
8-8	24-32	40-132	56-550
9-9	25-34	41-144	57-600
10-10	26-38	42-158	58-656
11-11	27-42	43-172	59-718
12-12	28-46	44-188	60-784
13-13	29-50	45-206	61-856
14-14	30-54	46-226	62-936
15-15	31-60	47-246	63-1024

With the exception of the entries with odd values, this table is identical to the table in [1,2] with the values doubled. The even code lines, which are used for 5 bit encoding, are all identical to the 2400 bps standard values. The altered values just provide interpolation values. (The unaltered 6 bit coding tables found in [1,2] will also work with slightly inferior interpolation.)

For 5 bit encoding and decoding, encode as above using only the even (code) entries in the tabl . To convert a 6 bit code to a 5 bit code, drop the least significant bit, and to convert a 5 bit code to a 6 bit code, double the code.

Appendix B
Log-Area Ratio Table

code	value	code	value	code	value	code	value	code	value
0	-.999999	52	-.876513	104	-.399851	156	.472724	208	.895815
1	-.979274	53	-.872270	105	-.384604	157	.486597	209	.899317
2	-.978521	54	-.867893	106	-.369144	158	.500230	210	.902708
3	-.977742	55	-.863376	107	-.353477	159	.513619	211	.905990
4	-.976934	56	-.858716	108	-.337609	160	.526762	212	.909167
5	-.976098	57	-.853910	109	-.321547	161	.539658	213	.912241
6	-.975232	58	-.848954	110	-.305298	162	.552306	214	.915216
7	-.974335	59	-.843844	111	-.288869	163	.564704	215	.918094
8	-.973405	60	-.838576	112	-.272267	164	.576853	216	.920879
9	-.972443	61	-.833146	113	-.255503	165	.588752	217	.923572
10	-.971446	62	-.827551	114	-.238583	166	.600401	218	.926178
11	-.970413	63	-.821787	115	-.221517	167	.611800	219	.928698
12	-.969344	64	-.815849	116	-.204315	168	.622951	220	.931135
13	-.968237	65	-.809734	117	-.186985	169	.633855	221	.933492
14	-.967091	66	-.803438	118	-.169538	170	.644512	222	.935770
15	-.965904	67	-.796957	119	-.151984	171	.654924	223	.937973
16	-.964675	68	-.790287	120	-.134334	172	.665094	224	.940103
17	-.963402	69	-.783424	121	-.116598	173	.675023	225	.942162
18	-.962084	70	-.776365	122	-.098787	174	.684713	226	.944152
19	-.960720	71	-.769106	123	-.080913	175	.694167	227	.946076
20	-.959308	72	-.761642	124	-.062987	176	.703388	228	.947935
21	-.957846	73	-.753971	125	-.045020	177	.712377	229	.949732
22	-.956333	74	-.746089	126	-.027024	178	.721139	230	.951468
23	-.954767	75	-.737992	127	-.009010	179	.729676	231	.953146
24	-.953146	76	-.729676	128	.009010	180	.737992	232	.954767
25	-.951468	77	-.721139	129	.027024	181	.746089	233	.956333
26	-.949732	78	-.712377	130	.045020	182	.753971	234	.957846
27	-.947935	79	-.703388	131	.062987	183	.761642	235	.959308
28	-.946076	80	-.694167	132	.080913	184	.769106	236	.960720
29	-.944152	81	-.684713	133	.098787	185	.776365	237	.962084
30	-.942162	82	-.675023	134	.116598	186	.783424	238	.963402
31	-.940103	83	-.665094	135	.134334	187	.790287	239	.964675
32	-.937973	84	-.654924	136	.151984	188	.796957	240	.965904
33	-.935770	85	-.644512	137	.169538	189	.803438	241	.967091
34	-.933492	86	-.633855	138	.186985	190	.809734	242	.968237
35	-.931135	87	-.622951	139	.204315	191	.815849	243	.969344
36	-.928698	88	-.611800	140	.221517	192	.821787	244	.970413
37	-.926178	89	-.600401	141	.238583	193	.827551	245	.971446
38	-.923572	90	-.588752	142	.255503	194	.833146	246	.972443
39	-.920879	91	-.576853	143	.272268	195	.838576	247	.973405
40	-.918094	92	-.564704	144	.288869	196	.843844	248	.974335
41	-.915216	93	-.552306	145	.305298	197	.848954	249	.975232
42	-.912241	94	-.539658	146	.321547	198	.853910	250	.976098
43	-.909167	95	-.526762	147	.337609	199	.858716	251	.976934
44	-.905990	96	-.513618	148	.353477	200	.863376	252	.977742
45	-.902708	97	-.500230	149	.369144	201	.867893	253	.978521
46	-.899317	98	-.486597	150	.384604	202	.872270	254	.979274
47	-.895815	99	-.472724	151	.399851	203	.876513	255	.980000
48	-.892197	100	-.458612	152	.414880	204	.880623		
49	-.888462	101	-.444265	153	.429686	205	.884605		
50	-.884605	102	-.429686	154	.444265	206	.888462		
51	-.880623	103	-.414880	155	.458612	207	.892197		

The preceding table is the log-area ratio (LAR) encoding-decoding table. For LAR encoding, if the K value is greater than or equal to the value entry and less than the next value entry, then the code is the current line. For decoding, use the code-to-value conversion as given. The slight bias in the decoding can be ignored, or if one wishes, an unbiased decoding can be achieved by using the average of the code line value and the next line value.

Appendix C

1200 bps Coding Tables for K4, K7, K8, and K9

The following tables are for encoding and decoding of several of the K values for the 1200 bps mode. All other K values are coded according to the government standard [1,2].

K4 frame-fill coding:

from	to	4 bit code value	decoded K
-.999999	-.699258	-8	-.738319
-.699257	-.621133	-7	-.660194
-.621132	-.543008	-6	-.582069
-.543007	-.464883	-5	-.503994
-.464882	-.386758	-4	-.425819
-.386757	-.308633	-3	-.347694
-.308632	-.230508	-2	-.269569
-.230507	-.152383	-1	-.191444
-.152382	-.074258	0	-.113319
-.074257	.014412	1	-.035194
.014413	.081992	2	.042931
.081993	.160117	3	.121056
.160118	.238242	4	.199181
.238243	.316367	5	.277306
.316368	.394492	6	.355431
.394493	.999999	7	.433556

K7 frame-fill coding:

from	to	3 bit code value	decoded K
-.999999	-.443358	-4	-.509757
-.443357	-.310558	-3	-.376957
-.310557	-.177758	-2	-.244157
-.177757	-.044958	-1	-.111357
-.044957	.087842	0	.021443
.087843	.220642	1	.154243
.220643	.353442	2	.287043
.353443	.999999	3	.419843

K8 frame-fill coding:

from	to	3 bit code value	decoded K
-.999999	-.519103	-4	-.586489
-.519102	-.384328	-3	-.451714
-.384327	-.249553	-2	-.316939
-.249552	-.114778	-1	-.182164
-.114777	.019997	0	-.047389
.019998	.154772	1	.087386
.154773	.289547	2	.222161
.289548	.999999	3	.356936

K9 frame-fill coding:

from	to	2 bit code value	decoded K
-.999999	-.246155	-2	-.363354
-.246154	-.011755	-1	-.128954
-.011754	.222645	0	.105446
.222646	.999999	1	.339846

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The following is a description of a low data rate voice terminal. The unit uses an LPC-10 algorithm which provides a 2400 bps NATO standard compatible mode and a 1200 bps frame-fill mode. The unit has an RS-449 digital interface with MIL-STD-188-114 levels. The user operates the unit from a telephone instrument with a PTT handset. The user interface models an augmented combination of an FM transceiver and a telephone to inform the user of the state of the data communications equipment (DCE) in a natural manner. These protocols cover both full and half duplex channels. The unit exhibits a DRT (intelligibility) score of 91.1% at 2400 bps and 88.7% 1200 bps in benign acoustic environments.

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